IN THE CLAIMS:

Please amend the claims as follows. The claims are in the format required by 35 C.F.R. § 1.121.

1. (Currently amended) A method comprising:

storing a plurality of sets of filter coefficients in a memory, wherein each set of filter coefficients defines a different filter function;

selecting a first one of the sets of filter coefficients which defines a first filter function;

interpolating the first selected set of filter coefficients; and

convolving the interpolated first selected filter coefficients with an input signal to produce a filtered output signal.

2. (Original) The method of claim 1, wherein the input signal comprises an audio signal.

3. (Original) The method of claim 2, wherein the input signal is convolved with the interpolated filter coefficients in a sample rate converter of a digital audio amplifier.

4. (Original) The method of claim 3, wherein the sample rate converter is implemented in a PWM amplifier.

5. (Original) The method of claim 1, wherein selecting the first one of the sets of filter coefficients comprises reading a value stored in a filter selection register and selecting the first one of the sets of filter coefficients based upon the value.

6. (Previously presented) The method of claim 5, further comprising changing the value in the filter selection register to a new value and selecting a new one of the sets of filter coefficients based upon the new value.

7. (Original) The method of claim 1, wherein the plurality of sets of filter coefficients are stored in a single memory.

8. (Original) The method of claim 1, wherein the first selected set of filter coefficients are interpolated according to a cubic spline algorithm.

9. (Original) The method of claim 1, wherein each of the plurality of sets of filter coefficients comprise polyphase filter coefficients.

10. (Currently amended) A system comprising:

a coefficient interpolator; and

a memory coupled to the coefficient interpolator;

wherein the memory is configured to store multiple sets of filter coefficients, wherein each set of filter coefficients defines a different filter function; and

wherein the coefficient interpolator is configured to interpolate a selected one of the sets of filter coefficients.

11. (Original) The system of claim 10, further comprising a convolution engine coupled to the coefficient interpolator and configured to convolve an input signal with interpolated coefficients corresponding to the selected one of the sets of filter coefficients to produce an output signal.

12. (Original) The system of claim 11, wherein the convolution engine is configured to convolve an audio input signal with the interpolated coefficients to produce an output audio signal.

13. (Original) The system of claim 12, wherein the convolution engine is implemented in a sample rate converter.

14. (Original) The system of claim 13, wherein the convolution engine is implemented in a pulse width modulation (PWM) amplifier.

15. (Original) The system of claim 10, further comprising a filter selection register configured to store a filter selection value, wherein the coefficient interpolator is configured to interpolate a set of filter coefficients indicated by the filter selection value in the filter selection register.

16. (Original) The system of claim 15, wherein the filter selection register is configured to allow modification of the filter selection value.

17-18. (Canceled)

19. (Original) The system of claim 10, wherein the memory comprises a single memory module configured to store the multiple sets of filter coefficients.

20. (Original) The system of claim 19, wherein each of the plurality of sets of filter coefficients comprise polyphase filter coefficients.

21. (Original) The system of claim 10, wherein the coefficient interpolator is configured to interpolate the selected set of filter coefficients according to a cubic spline algorithm.

22. (Currently amended) A method comprising:

storing a plurality of sets of filter coefficients in a memory, wherein each set of filter coefficients defines a different filter function;

selecting only one of the sets of filter coefficients;

interpolating the selected set of filter coefficients; and

convolving the interpolated set of filter coefficients with an input signal to produce a filtered output signal.

23. (Previously presented) The method of claim 22, further comprising performing the method in a sample rate converter of a digital PWM amplifier, wherein the input signal comprises an audio signal.